

Error Probability in Redundant Packet Sending over IP Network

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Abstract. *In this paper we calculate error probability of packetized signal when method of redundant packet sending is used in IP network. The number of repeated signaling packets from each interval of packetization is determined to achieve the desired error probability. The method for management of this number of repetitions is developed based on the new analysis. This method is especially important in the case of sending signaling criteria of classic telephony network over IP network, because it makes possible to reach the same error probability as in classic telephony network.*

Keywords

IP network, error probability, redundant packet.

1. Introduction

Today IP network has become the dominant network in communication. It contributes to the usage of new telephony services and, in the same time, the existing services must be performed differently. For example, the classic telephony services are performed using IP network. When these classic telephony services are realized by IP network, it is necessary to achieve the service quality at least as it was in classic telephony network. This is not often realized easily, because IP network is under the influence of many different impacts (speech signal delay, echo, jitter, speech signal compression, packet loss, transcoding), which reduce the quality of one telephone connection. This paper deals with the error probability in message packet sending, and with redundant packet coding, as one of the methods for error probability decrease in message transmission. We calculate the error probability for redundant packet coding and the required packet repetition rate to achieve desirable error rate. After that we analyze the capabilities of Real Time Control Protocol (RTCP) and present the method for management of repetition rate in the process of packet loss control.

2. The Structure of Internet Packet

For one telephone connection realized by IP network,

as for each telephone connection, it is necessary to transfer signaling messages for call setup and release, and the coded speech signal between subscribers. The errors can appear during the transmission of these signaling. In order to achieve the appropriate quality of the transmitted signal, which is under the influence of transmission errors, two methods are used. The first method is the retransmission of incorrectly transmitted messages. There are two possible methods of retransmission: the first one is negative acknowledgement from the receiving side that the message is not transmitted correctly, and the second one is the non-existence of the acknowledgement that the message is received within prescribed time interval. These methods can lead to the significant delay in message transfer [1]. The implication is that, because of the problems with delay, the time relation between generated and reproduced signals can not be adequate.

The second method for the reduction of error effect in message transmission is based on the Real Time Protocol (RTP) [2], [3]. Internet packets, constructed using RTP, can contain structures, which enable the more precise signal reconstruction on the receiving side. So, better signal quality is achieved without the signal retransmission. One possible way for this is redundant packet sending.

The process of generating redundant message packets can be explained by Fig. 1.

Fig. 1a) presents digitalized signal samples. Their content is prepared for packetization, as it is symbolically presented in Fig. 1b). The packet can contain digitalized signal samples, or it is possible to make some signal processing first (for example, the detection of signaling criteria). Fig. 1c) presents the structure of redundant packets.

According to Fig. 1c), each packet consists of the content from the last packetization interval and from some previous packetization intervals (3 intervals). For example, the packet generated after the interval 4, contains packetized data from intervals 1, 2, 3 and 4; the packet generated after the interval 5, contains packetized data from intervals 2, 3, 4 and 5, and so on. It can be said that each packet contains data from k packetization intervals and each packetized data is sent over IP network totally k times.

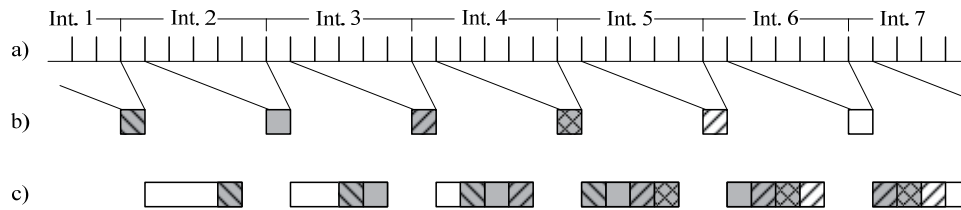


Fig. 1. The process of generating redundant message packets.

Ether. Preamble (8 bytes)	Ether. Header (14 bytes)	IP Header (20 bytes)	UDP Header (8 bytes)	RTP Header (12 bytes)	RTP payload	CRC Check (4 bytes)
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Fig. 2. The process of generating redundant message packets.

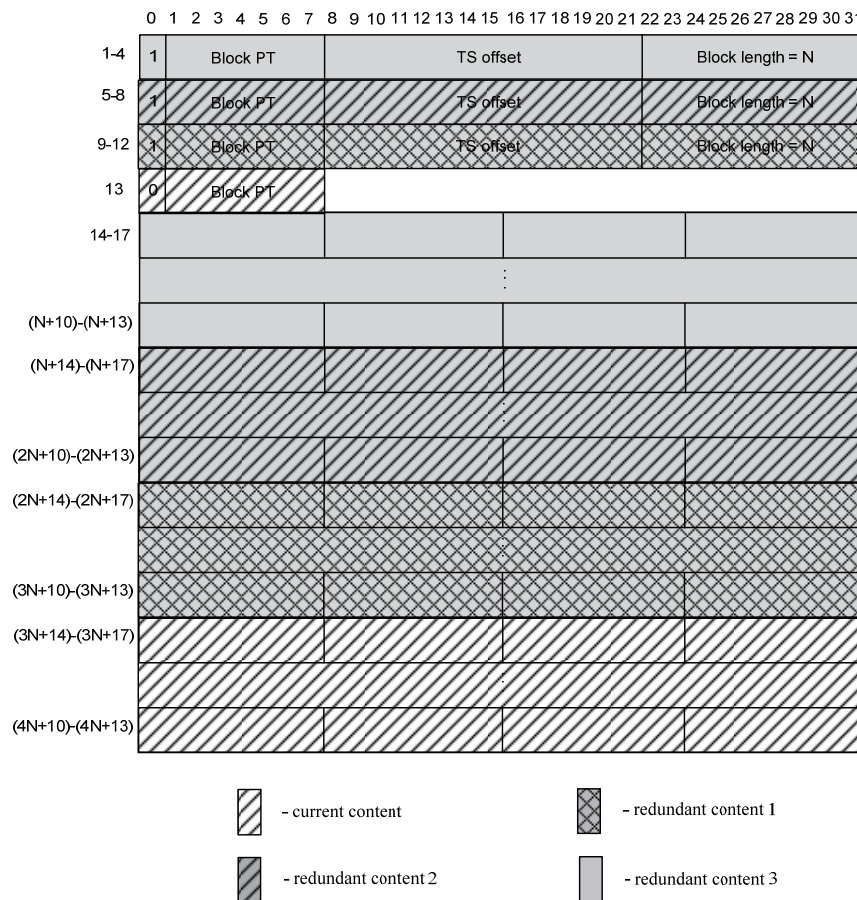


Fig. 3. The RTP payload structure.

The structure of one Internet packet is presented in Fig. 2 [4]. All headers in the packet (Ethernet, IP, UDP and RTP) and their length in bytes are emphasized. At the packet beginning Ethernet Preamble is situated, and at the end CRC packet check exists. All these parts have the fixed length. Besides this, in packet exists the part, which carries the content of the message (RTP *payload*), and its length depends on the content and type of the message.

The structure of the RTP payload in one Internet packet is presented in Fig. 3 [2], [3]. The bytes in RTP

payload of Internet packet are numerated by numbers 1 to $13+4N$. The packet in Fig. 3 contains data from the last packetization interval and from previous 3 packetization intervals.

The first 13 bytes in the packet define the part of the packet, which presents the structure of the transmitted signal. For each of the signals from three previous packetization intervals four bytes are reserved (bytes 1 to 4 for the interval, which is located three packetization interval before the current packetization interval, and so on). For the

current packetization interval only one byte is reserved. The content for three previous packetization intervals starts with the bit value 1 in the first byte, which means that the data is not intended for the last packetization interval in the packet. The remaining 7 bits in the first byte present the type of the coded signal (Block payload type (PT)). (In our case all contents are of the same type, but it is not necessary to be fulfilled in all situations). The remaining three bytes define the beginning time point of the transmitted signal in redundant packet part relative to the beginning of the current (last) transmitted signal in that packet, and the block length of the part coding the transmitted signal (in our case, these blocks for all signal parts have the length N bytes). For the last transmitted signal, only one byte is reserved (byte number 13). This byte contains the bit indicating that this is the last signal part in this packet (bit value 0) and the block PT.

As it is said, the data for each packetization interval contain N bytes. Bytes 14 to $13+N$ are connected with the packetization interval, which is located 3 time intervals before the current one, and so on, until the bytes $14+3N$ do $13+4N$, which are connected with the current packetization interval.

3. Error Probability in Message Packet Sending over IP network

Error probability in message packet sending over IP network can be expressed as:

$$P_g = 1 - P_{IP} = 1 - (1 - BER)^n \quad (1)$$

where

- P_{IP} – the probability of correct message sending;
- BER – Bit Error Rate: error probability of one bit sending in Internet message;
- n – number of bits in Internet message.

For the small values of BER , which are real in telecommunication today, the error probability can be expressed approximately as:

$$P_g \approx n \cdot BER = (L_{HEADER} + L_{RTPPAY}) \cdot BER \quad (2)$$

where:

- L_{HEADER} – length of all message headers;
- L_{RTPPAY} – payload length in RTP part of the packet.

According to Fig. 2, L_{HEADER} includes preamble, Ethernet, IP, UDP, RTP header, and CRC check of packet content. The length of these fields is constant and, according to Fig. 2, is 66 bytes.

The length L_{RTPPAY} includes payload presented in Fig. 3. This length is variable and depends on the structure of the content, which is packetized, i.e. on the length of the packetization interval for each packet, and on the number

of packetization intervals, which are included in one redundant message packet.

Considering the example in Fig. 3, where we present the RTP payload in the case when the packet consists of the data from the last packetization interval and from the previous three intervals, the total number of bits in RTP payload can be calculated in the general case, when it contains data from k packetization intervals. This number of bits is:

$$L_{RTPPAY} = 8 \cdot (4 \cdot (k-1) + 1 + N \cdot k). \quad (3)$$

Replacing the value from (3) in (2), we obtain:

$$P_g \approx (504 + 32 \cdot k + 8 \cdot N \cdot k) \cdot BER. \quad (4)$$

Formula (4) determines the error probability for only one packet, containing the data from k packetization intervals. The packetized signal from one packetization interval will be lost if k consecutive packets are not received at the receiving side of the connection. According to this, the content from one packetization interval will be lost with the probability:

$$P_g \approx ((504 + 32 \cdot k + 8 \cdot N \cdot k) \cdot BER)^k. \quad (5)$$

For the calculation of the number of content repetition k , we used the program Mathematica. The expression (5) is, first, presented in its logarithmic form:

$$\log P_g \approx k \cdot \log((504 + 32 \cdot k + 8 \cdot N \cdot k) \cdot BER) \quad (6)$$

and the graphics in Fig. 4 are obtained. In Fig. 4 the error probability of one bit is $BER = 10^{-5}$ (Fig. 4a), $BER = 10^{-6}$ (Fig. 4b) and $BER = 10^{-7}$ (Fig. 4c).

The number k of repeated signaling packets will be the first integer number, which satisfies the error probability in message packet sending.

The desired value of error probability depends on the transmitted data type. For example, if speech packets are transmitted, the error probability $P_g \leq 10^{-2}$ is acceptable [5]. In the case of signaling criteria transmission, it is necessary to achieve far less error probability. When signaling messages of signaling system 7 (SS7) are transmitted, the necessary error probability is 10^{-10} [6], and we can take this value as the desired error probability when signaling criteria is transmitted.

In Fig. 5 the error probability is presented in the function of the number of repeating the redundant part of the message. The graphs are presented for $BER = 10^{-8}$ (Fig. 5a) and $BER = 10^{-6}$ (Fig. 5b). In both cases value $P_g = 10^{-2}$ can be achieved for $k = 1$, which means that there is no need to repeat the content of packetized signal in the case of voice signal transmission. The value $P_g = 10^{-10}$ can be achieved for $k = 2$ in the first case and for $k = 4$ in the second case, which means that it is necessary to repeat the content of the packetized signal in the case of signaling criteria transmission.

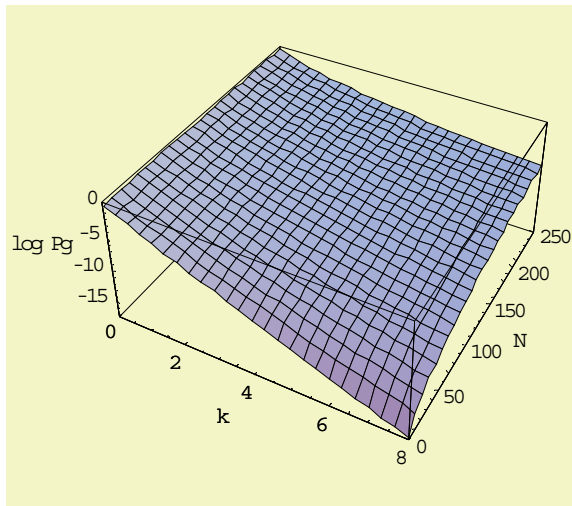
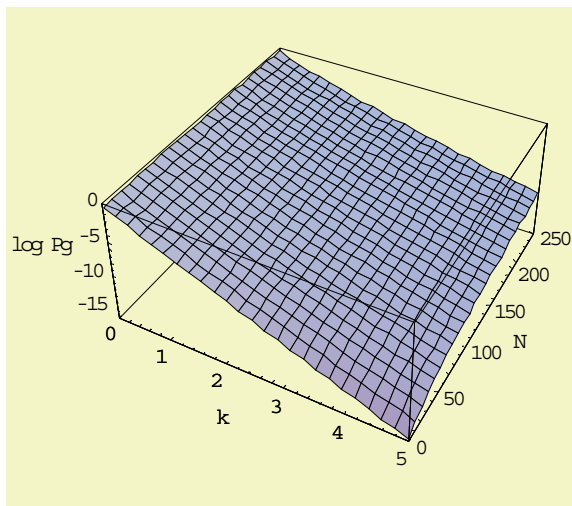
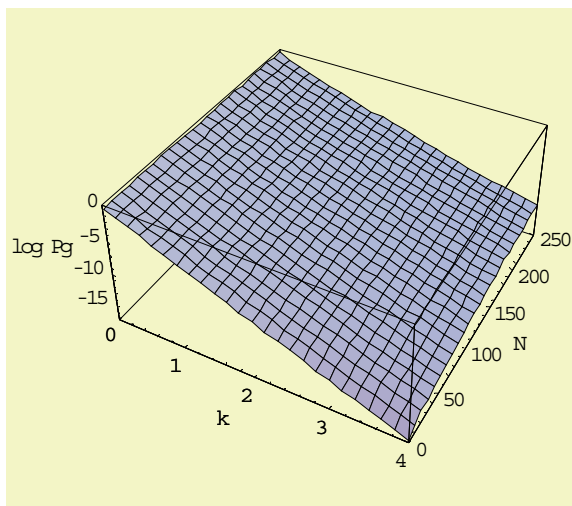
a) $BER = 10^{-5}$ b) $BER = 10^{-6}$ c) $BER = 10^{-7}$

Fig. 4. Error probability of message transmission in the function of the redundant content length and of number of repeated signaling packets.

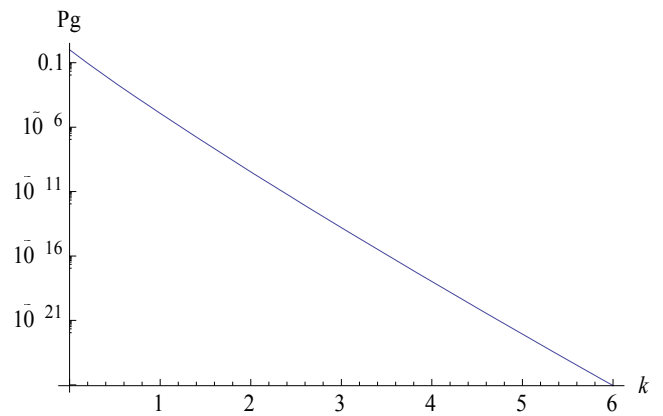
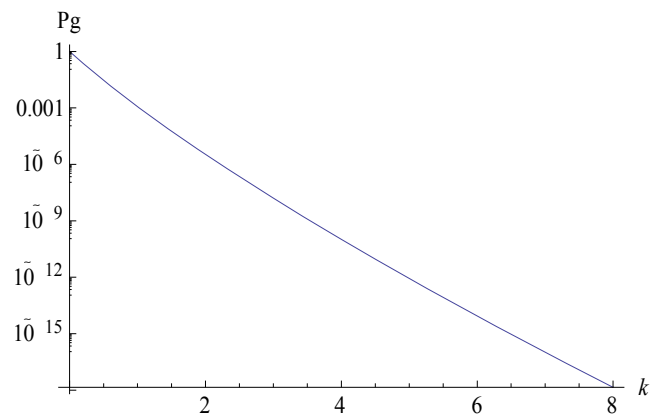
a) $BER = 10^{-8}, N=80$ b) $BER = 10^{-6}, N=80$

Fig. 5. Error probability of message transmission in the function of the number of repeating redundant content for $N=80$ bytes in each packetization interval

4. Practical Management of Error Probability in Message Packet Sending

Speech and signaling message packets must be sent in real time. This means that retransmission of incorrectly transmitted messages cannot be used. That's why it is very important to control the packet correctness at the receiving side. The receiving side analyses all important data for packet sending over IP network (calculates percent of lost packets, percent of packets which had to be eliminated, jitter in packet sending, burstiness in packet loss, quality of transmitted signal, etc.). Based on these collected data, the receiving side sends the report about the transmitting quality to the sending side, and the sending side can correct the process of message sending to achieve the desired quality.

Fig. 6 presents one of the packets, intended for sending the report about message packet sending. This report is based on RTCP usage [7]. Each row in this figure contains one 32-bit word. First eight bytes present the header of RTCP packet, and the remaining part is the packet payload.

The first byte in the payload is the block type (BT), which defines the remaining part of the packet. The part marked as “Block length = 8” identifies that the remaining packet length is 8 32-bit words. The record contains significant amount of data, and for us the most important are probability of packet loss (loss rate – *LR*) and the probability of packet elimination (discard rate – *DR*). The time interval between RTCP packets sending is defined according to [8]. Its upper bound is limited by the fact that RTCP bandwidth is recommended to be 5% of the session bandwidth. In fact, the higher the number of content repetition *k*, the higher RTCP bandwidth possible and, so, the lower time interval between RTCP packets.

Let us suppose that packet is lost (or discarded), because the error happened on one bit in packet. In that case the corresponding *BER* can be calculated as:

$$BER = \frac{P_g}{504 + 32 \cdot k + 8 \cdot N \cdot k} \quad (7)$$

and this formula is used for error calculation in preparing RTCP messages.

The repetition of packet content causes the increased required bandwidth for message transfer and the increased needed buffer capacity in intermediate switching elements. These are the disadvantages of used method for error probability decreasing. But, as the quality of speech signal is less sensitive on errors in packet transmission than signaling criteria sending, it is possible to use this method only for signaling criteria transmission, as explained according to the examples in previous section. Thus we reduce the possibility of buffer overflow in the intermediate switching elements, because the time duration of signaling periods is, usually, much less than time duration of speech periods.

0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
								PT=207								Total length															
BT=7								Reserved bits								Block length = 8															
Source address																															
Loss rate								Discard rate																							
Remaining 6 32-bit words																															

Fig. 6. Packet with the report about message transmission.

5. Necessary Number of Repeating the Content in Redundant Packets

After each report about the probability of incorrectly transmitted and the probability of discarded message packets, which the transmitting side receives from the receiving side in one Internet connection, the necessary number of repeating one content in redundant packets *k* is calculated in order to achieve the desired error probability in the next time interval. The procedure carried out in that case is as follows:

- *BER* calculation using the error probability in packet sending and the formula (7);
- based on the obtained *BER*, calculation of the corresponding loss probability (*P_g*) of content from one packetization interval when redundant sending is used. This is the probability to lose *k* consecutive packets. For this calculation, formula (5) is used;
- comparison of obtained *P_g* with the desired value;
- if the obtained value *P_g* is greater than the desired value, determination of the greater value *k* and the verification by the formula (5) is it possible with the

new value *k* to achieve the desired value *P_g*. The value *k* is increased until the value lower than the required one is obtained;

- if the obtained value *P_g* is less than the desired, it is calculated if some lower value *k* also satisfies for the required error probability.

6. Conclusion

In this paper we present the method for error probability improvement when signals of classic telephony (speech and, especially, signaling criteria) are sent over IP network. The method is based on management of the number *k* of redundant message packet repeating. The criterion for the number of repetition is the desired error probability in message transmission. The RTCP packets are used to manage the repetition number of each part of packetized signal in redundant packets when the probability of losing the content from one packetization interval is about 10⁻¹⁰. This error probability is demanded in SS7 signaling criteria sending. It has to be achieved also when signaling criteria are transmitted over IP network.

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